Chapter 6

Packet Scheduling and Buffer Management in HSDPA

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Contents

6.1 Introduction

Providing high-speed data has always been an important goal of the wireless community. The 3rd Generation Partnership Project (3GPP) has introduced the High-Speed Downlink Packet Access (HSDPA) as a step forward in this direction. HSDPA evolved from WCDMA (Wideband Code Division Multiple Access) utilizing a number of existing technologies. Several techniques have been employed to compensate for the changing link conditions. The main theme is based on link adaptation by modifying the transmission parameters of the system to adapt to the instantaneous transmission conditions. Among many, HSDPA employs fixed spreading factor, adaptive modulation and coding (AMC), fast scheduling, and physical layer retransmission by applying Hybrid Automatic Repeat Request (HARQ) to provide high-speed downlink packet access by means of High-Speed Downlink Shared Channel (HS-DSCH). All this implies that substantial changes have been made to the Node B to enhance Release 99 WCDMA with packet scheduling embedded in a new MAC sub-layer known as the MAC-hs (Medium Access Control) (MAC-high speed). Because this necessitates data buffering at the air interface, which poses a bottleneck to end-to-end communication, buffer management in the Node B MAC-hs is essential and critical for Quality-of-Service (QoS) provision.

In Node B, an efficient packet scheduling mechanism is crucial to HSDPA performance. Hence, this chapter discusses various packet scheduling algorithms that have been proposed for HSDPA. Packet scheduling algorithms that support multimedia traffic with diverse concurrent classes of flows being transmitted to the same end user are discussed. Furthermore, new approaches for such end-user multimedia sessions based on integrated

packet scheduling with buffer management are also presented. In these approaches, the packet scheduling functionality selects a user for downlink transmission based on a given scheduling discipline (i.e., inter-user prioritization), while the buffer management scheme determines the class of flow to be transmitted from the users' multiplexed flows (i.e., inter-class prioritization).

The chapter begins with a detailed description of the HSDPA physical and MAC layers as given in the 3GPP standards, followed by a discussion of the HSDPA packet scheduling algorithms. Finally, the integrated buffer management and packet scheduling solutions for HSDPA multimedia sessions with concurrent diverse flows are presented.

6.2 HSDPA Physical Layer

On the radio system level, several WCDMA functionalities are adapted to enable HSDPA. Three of the fundamental properties of WCDMA have been disabled: soft handover, variable spreading factor (SF), and fast power control. In HSDPA, only hard handover is allowed. The SF is fixed at SF16 under one scrambling code [1]. Figure 6.1 depicts the spreading factor and the number of channelization codes available for HSDPA. Fifteen channel codes are used for data transmission. One code is used for signaling with an SF of 128. AMC techniques have been implemented. Depending on the channel condition, the scheduler can have a choice between QPSK (quadrature phase-shift keying) and 16QAM (quadrature amplitude modulation). In QPSK, each signaling element represents 2 bits; however, in 16QAM, each signaling element represents 4 bits, which in theory can double the amount of data carried in a frame. Turbo coding is employed in HSDPA, ranging from $R = 1/4$ to $R = 3/4$, depending on the channel condition. Spreading is applied to the physical channel after coding and modulation. Spreading, in effect, increases the bandwidth of the signal. The number of chips per data symbol is called the spreading factor. A scrambling operation is applied to the signal, and this operation is a means to separate base stations [2].

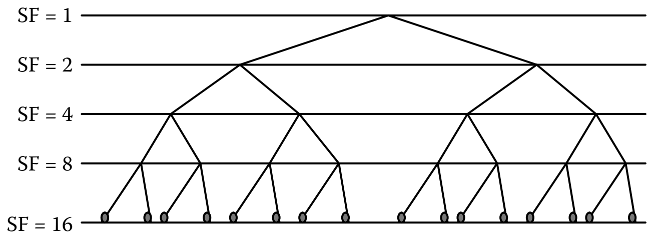

Figure 6.1 Spreading factor.

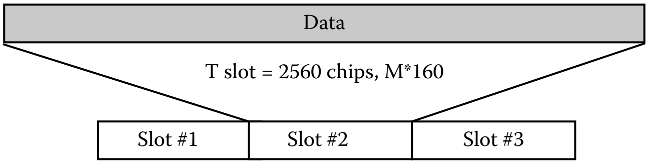

Figure 6.2 Subframe structure for the HS-PDSCH.

Three new physical channels have been defined for HSDPA: the HS-PDSCH (High-Speed Physical Downlink Shared Channel) used to carry data on the downlink direction, the HS-SCCH (High-Speed Shared Control Channel) used to carry signaling in the downlink direction, and the HS-DPCCH (High-Speed Dedicated Physical Control Channel) used to carry signaling in the uplink direction. An HS-PDSCH corresponds to one channel code of fixed spreading factor from the set of 15 channelization codes available for HS-DSCH transmission. Multi-code transmission is allowed, which translates to a UE (user equipment) being assigned multiple codes in the same HS-PDSCH transmission time interval (TTI = 2 ms), depending on the UE capability. The subframe and slot structure of HS-PDSCH are shown in Figure 6.2.

HS-PDSCH can use QPSK or 16QAM modulation symbols, where M is the number of bits per modulation symbols; that is, $M = 2$ for QPSK and $M = 4$ for 16QAM. The 10-ms frame of WCDMA has been divided into five subframes, 2 ms each for fast scheduling and retransmission. The HS-SCCH is a fixed-rate (60 kbps, SF = 128) downlink physical channel used to carry downlink signaling related to HS-DSCH transmission. Figure 6.3 shows the subframe structure of the HS-SCCH.

HS-SCCH is transmitted two time slots ahead of HS-PDSCH. It carries critical signaling information such as the channelization code set, modulation scheme, and transport block size to the UE. HS-SCCH also carries the HARQ process number, redundancy version, and UE identity, among other information [3].

HS-DPCCH is used to carry the signaling information in the uplink direction. The channel has SF = 256, 10 bits per slot, and 30 bits for the subframe

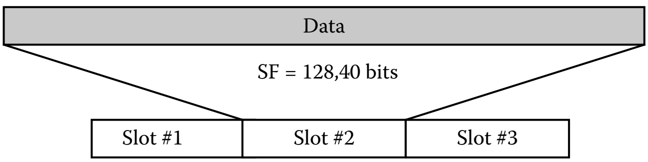

Figure 6.3 Subframe structure for the HS-SCCH.

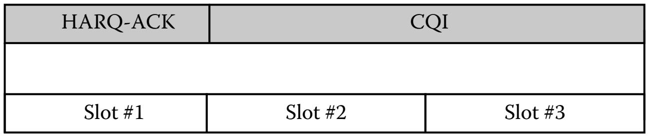

Figure 6.4 Subframe structure for the HS-DPCCH.

(15 kbps). There is, at most, one HS-DPCCH per UE. HS-DPCCH carries the ARQ ACK, in one slot and the Channel Quality Indicator (CQI) in the other two slots. Figure 6.4 shows the subframe structure for the HS-DPCCH.

The CQI represented by 5 bits for a value is between 1 and 30; based on this feedback from UE, the scheduler in Node B decides on the modulation and transport block size to send in the next TTI transmission if there is available data for the UE in the buffer [4]. HS-DPCCH is transmitted 7.5 time slots (5 ms) after the reception of the HS-PDSCH.

6.3 HSDPA MAC Architecture

To support HS-DSCH, a functional split in the MAC is required. This new entity is called MAC-hs and it is deployed closer to the radio link in Node B for fast retransmission. In each cell that supports HS-DSCH, there is one MAC-hs entity, responsible for management of physical resources allocated to HSDPA. HSDPA did not affect the architecture of the upper layers. MAC-hs is composed of four functional entities as depicted in Figure 6.5 [5]. These entities are described below.

6.3.1 Flow Control

A flow control function is needed to govern the exchange of data between MAC-hs and MAC-d. There are two cases for implementing flow control functionality. In the first case, an SRNC (serving radio network controller) is connected to Node B through the Iub interface. MAC-d entity is located in the SRNC and connected to MAC-hs through the Iub interface. The SRNC in the second case is connected to the CRNC (controlling RNC) through the Iur interface and, in turn, CRNC is connected to Node B through the Iub interface. In the first case, the flow control is directly exchanging data between MAC-d (located in the SRNC) and MAC-hs (located in Node B). In the second case, there are two flow controls, one between MAC-d (located in SRNC) and MAC-c/sh (located in CRNC) and the second one between MAC-c/sh and MAC-hs. The UTRAN side of the MAC architecture is depicted in Figure 6.6.

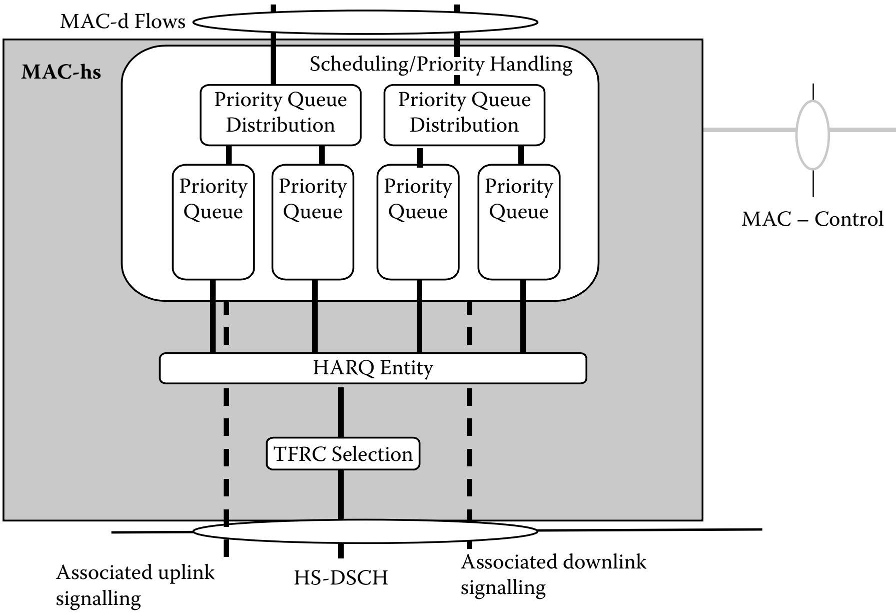

Figure 6.5 MAC-hs UTRAN side. (*Source:* From 3GPP, Technical Specifications Group RAN, Medium Access Control (MAC) Protocol Specifications, 3GPP TS 25.321 version 8.4.0.)

There are two control messages available in the 3GPP standards for exchanging data between MAC-hs and MAC-d. HS-DSCH Capacity Request is shown in Figure 6.7. Entity A and Entity B can be either SRNC and Node B, respectively, if the flow control is directly between the SRNC and Node B, or (SRNC, DRNC) and (CRNC, Node B) if the flow control is handled separately on Iur and Iub. The capacity request message is sent from SRNC/DRNC to Node B to request that data be transferred to Node B and to indicate the buffer size of the user.

The HS-DSCH Capacity Allocation signaling message is illustrated in Figure 6.8. This message indicates to SRNC/DRNC the allocated buffer space in Node B for a specific user.

SRNC indicates the amount of data available for a user-given Common Transport Channel Priority Indicator (CmCH-PI). The range of the priority is between 0 and 15, with 0 being lowest priority and 15 being the highest priority. SRNC also indicates the buffer size of the user—that is, the amount of data in the SRNC buffer. In the capacity allocation control frame, the amount of data or credits that SRNC is allowed to send is specified by Node B. In case of congestion on the Iub interface, the credits are reduced equally for all users [6].

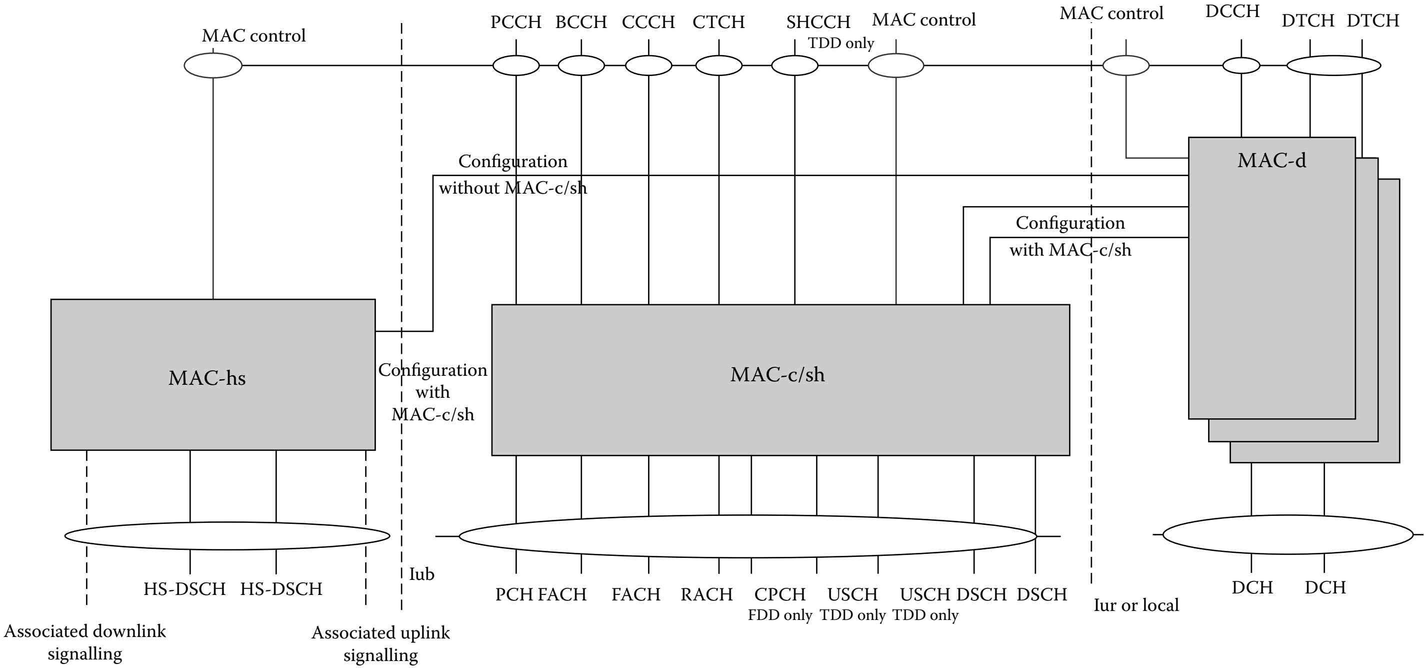

Figure 6.6 UTRAN side of the MAC architecture. (*Source:* From 3GPP, Technical Specifications Group RAN, Medium Access Control (MAC) Protocol Specifications, 3GPP TS 25.321 version 8.4.0.)

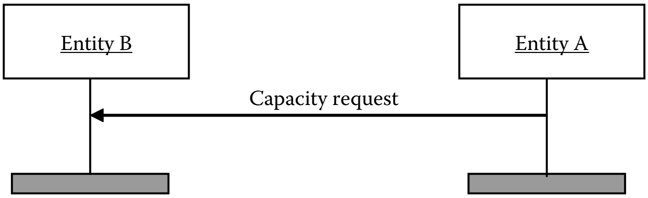

Figure 6.7 HS-DSCH capacity request.

Flow control is intended to limit layer 2 signaling latency and reduce discarding and retransmission of data as a result of HSDPA congestion. Flow control is provided independently by MAC-d flow for a given MAC-hs entity. As a matter of fact, one of the design goals of MAC-hs is to limit signaling to the upper layers.

Studies on HSDPA performance (see, for example, [8–10]) indicate that Iub flow control has a significant impact on MAC-hs packet scheduler performance and the resulting end-to-end QoS (Quality of Service). In [11], a flow control algorithm to support transfer of multiplexed diverse flows in an end-user multimedia session is proposed for the integrated buffer management and scheduling approach that is discussed later.

6.3.2 Scheduling and Priority Handling

This function manages HS-DSCH resources between HARQ entities and the data flows according to their priority. There is one priority queue entity (HARQ entity) per user, and there are eight different queues in each entity. For each MAC-hs PDU (protocol data unit) being serviced, the priority entity determines the queue ID and the transmission sequence number (TSN is explained when we discuss the frame format of MAC-hs PDU). Based on the HS-DPCCH report in the uplink signaling, MAC-hs has the choice of new transmission or retransmission of lost PDU. 3GPP standards do not specify

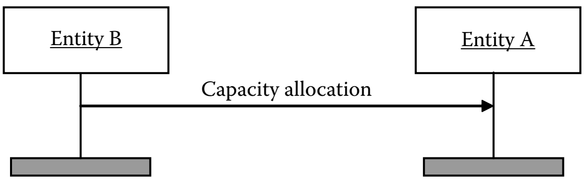

Figure 6.8 HS-DSCH capacity allocation.

the scheduling technique to use; however, several scheduling techniques that have been researched are described later in this chapter.

6.3.3 HARQ

There is one HARQ entity for each user. Each entity can support multiple instances of HARQ processes. There is one HARQ process created per HS-DSCH per TTI. There are two main schemes of implementing HARQ, *Chase Combining* and *Incremental Redundancy*. Chase Combining is retransmission of the same data block. In Incremental Redundancy, each retransmission includes additional redundancy bits from the channel encoder (HARQ type 2). HARQ type 3 is also an incremental redundancy scheme but with a difference that each transmission is self-decoded. There are two types of protocols that are used with HARQ: Selective Repeat (SR) and Stop-and-Wait (SAW). The first protocol repeats the blocks that are in error and is usually insensitive to delay. However, the disadvantage of this scheme is that the UE memory requirements very highly depends on the maximum number of blocks in the transmission sequence. The other disadvantage is that HARQ requires the receiver to reliably determine the sequence number of the transmission; therefore, stronger code must be used for the sequence number. The SAW protocol is simple and requires very little overhead. It keeps working on the current block until it is received successfully. This protocol will save signaling bandwidth and reduce the amount of memory required by the UE. However, such a simplistic approach comes at the expense of waiting of 7.5 time slots (5 ms) for the acknowledgment to arrive at the transmitter. This wait will reduce the system capacity. Therefore, we run several instances in parallel with the SAW to eliminate this problem and fully utilize the system resources [1].

6.3.4 Transport Format and Resource Combination (TFRC)

Depending on the reported channel quality condition, TFRC is determined. There are two values for modulation (QPSK and 16QAM) and several coding values ranging from 1/4 to 3/4. The combination of the modulation and coding is adjusted based on the channel quality conditions as previously mentioned. TFRC largely depends on the UE capabilities, modulation, and coding rate (Table 6.1) [5, 7].

6.3.5 MAC-hs Protocol Data Unit

MAC-hs PDU is a bit string but not necessarily in multiples of 8 bits. It consists of a MAC-hs header and one or more MAC-hs SDUs (service data

Table 6.1 Theoretical Bit Rates with 15 Multi-codes

TFRC	*Modulation*	*Effective Code Rate*	*Maximum Throughput (Mbps)*
1	QPSK	1/4	1.8
2	QPSK	2/4	3.6
3	QPSK	3/4	5.3
4	16QAM	2/4	7.2
5	16QAM	3/4	10.7

units). The MAC-hs header is of variable size. Each MAC-hs SDU equals one MAC-d PDU. A maximum of one MAC-hs PDU can be transmitted in a TTI per UE. The MAC-hs SDUs belong to the same queue. MAC-hs consists of the following fields (Figure 6.9):

- Version Flag (VF): This is a 1-bit flag providing extension capabilities to the PDU format. This bit is always set to 0 and the value 1 is reserved for future use.
- Queue Identifier (Queue ID): This is a 3-bit field that provides identifications to the reordering queue in the receiver, in order to support independent buffer handling of data belonging to different reordering queues.
- Transmission Sequence Number (TSN): This field is used to support in-sequence delivery to higher layers. The TSN identifies the transmission sequence on HS-DSCH. This field is 6 bits.
- Size Index Identifier (SID): This field identifies the size of a set of consecutive MAC-d PDUs. The MAC-d PDU size for a given SID is configured by higher layers and is independent for each queue. This field is 3 bits long.

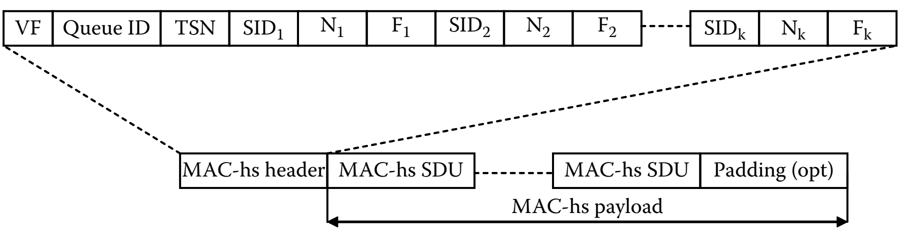

Figure 6.9 MAC-hs PDU structure.

- Number of MAC-d PDUs (N): This is the number of consecutive MAC-d PDUs with equal size. This field is 7 bits long. In FDD (frequency division duplex) mode, the maximum number transmitted in one TTI is assumed to be 70.
- Flag (F): This is a 1-bit field. If the value is 0, then the field is followed by an additional SID; if the value is 1, then the field is followed by a MAC-d PDU. The maximum number of repeated $F = 0$ in a TTI is assumed to be 7.

3GPP standards do not specify a certain scheduling mechanism for HSDPA. There are three QoS parameters specified in the Iub: the guaranteed bit rate (GBR), the Scheduling Priority Indicator (SPI) of values 0 to 15 as mentioned earlier, and the discard timer (DT), which specifies how long the packet should be buffered in Node B (MAC-hs) before it is discarded. The mapping of these parameters and how they are used in MAC-hs scheduler has been left to the operators. The scheduler can schedule all UEs within a cell based on the SPI from the higher layer (in the RNC). Alternatively, the UEs can be scheduled based on other metrics specified according to a given scheduling algorithm in the MAC-hs; the most popular algorithms are discussed later. The scheduler determines the HARQ entity and the queue to be serviced. It sets the TSN, starting from value 0 for each Queue ID. The scheduler also increments the value by one for each new transmission, but not for retransmission. It indicates the Queue ID and TSN to the HARQ entity for each MAC-hs PDU to be transmitted. The scheduler may decide to discard any out-of-date MAC-hs SDU. The scheduler also determines the redundancy version for the transmitted and retransmitted frame. The next section discusses well-known packet scheduling algorithms that can be implemented for scheduling the UEs in the MAC-hs packet scheduler.

6.4 HSDPA Packet Scheduling Algorithms

The MAC-hs packet scheduler is responsible for accommodating, within the limited and variable resources, HSDPA users with different QoS requirements and those experiencing different air interface conditions [12]. There are several possible solutions to such a scheduling problem, and this section discusses various algorithms proposed by researchers for HSDPA MAC-hs packet scheduling, including techniques that support multiple classes of flows per user session.

With the purpose of enhancing the cell throughput (capacity), the HSDPA scheduling algorithm can take advantage of the instantaneous channel variations and temporarily raise the priority of the users. Because the users' channel quality varies asynchronously, the time-shared nature of HS-DSCH

introduces a form of selection diversity with important benefits for spectral efficiency [13]. However, this could mean that users more distant from Node B (at cell edge), and therefore requesting lower data rates, could be starved for service. Consequently, the scheduling algorithms must balance the conflicting goals of maximizing throughput, while at the same time ensuring some degree of fairness to all users requesting a service [14].

Opportunistic scheduling methods or *channel-aware schedulers* are those that exploit multi-user (selection) diversity, such that with a large number of users in the cell, a gain in total cell throughput can be achieved by giving transmission priority to users with favorable channel conditions over those with poor channel conditions. On the other hand, *blind* scheduling methods do not take into consideration the radio conditions of the users. According to [15], with opportunistic scheduling, the QoS of all users can be improved over (blind) scheduling schemes that do not take channel conditions into account. Gutierrez [13] classifies the well-known HSDPA packet scheduling algorithms according to the pace of the scheduling process into *fast* and *slow* as described below.

6.4.1 Fast Scheduling Methods

Fast scheduling methods are those that base the scheduling decisions on the recent UE channel quality measurement so that the instantaneous variations in the user's supportable data rate can be tracked. With fast scheduling methods, the scheduling decisions are executed on a TTI basis. Where there is a sufficient number of time multiplexed users, fast scheduling methods can exploit multi-user selection diversity to provide significant capacity gain. Examples of fast scheduling methods include Maximum C/I, Proportional Fair, and Fast Fair Throughput.

6.4.1.1 Maximum C/I Scheduling

The Maximum Carrier-to-Interference Ratio scheduler (Max-C/I), which is also referred to as the maximum throughput scheduler, is designed to maximize HSDPA cell throughput. The Max-C/I packet scheduler always schedules the user with the best instantaneous channel quality. Because the Max-C/I scheduler monopolizes the cell resources for users with favorable channel conditions, there may be a number of users at the cell edge that may never be scheduled. However, because fast fading dynamics have a larger range than the average radio propagation conditions, it is possible for users with poor average radio conditions to still have access to the shared channel. Max-C/I is inherently unfair in scheduling resources and suffers from coverage limitations, which are its main drawbacks.

6.4.1.2 Proportional Fair Scheduling

This scheduling algorithm serves the user with the best *relative* channel quality according to the following scheduling metric:

$$P_i = \frac{r_i(t)}{R_i} \quad i = 1, \ldots, K$$

where P_i denotes the user priority, r_i(t) is the instantaneous data rate that can be supported by the user in the current TTI if served by the packet scheduler, and R_i is the average throughput experienced by user i. The denominator in the scheduling metric allows a user that is being allocated little scheduling resources to increase its priority over time. R_i can be updated every TTI prior to selecting a user by employing the following first-order recursion:

$$R_i(n) = (1-\beta)R_i(n-1) + \beta r_i(n-1)$$

where r_i $(n-1)$ is the requested rate of user i in the previous TTI $(n-1)$ if the user i was selected for transmission in TTI $(n-1)$, and 0 otherwise. Because β is the time constant of the (exponential smoothing) filter, β^{-1} equals the equivalent averaging period in TTIs. Thus, Kolding [16] recommends that the averaging length should be set long enough to ensure that the process averages out the fast fading variations, but short enough to still reflect medium-term conditions such as shadow fading. Note that the throughput calculation for a user is only done for periods of time when the user has buffered data in the MAC-hs. This is important for the stability of QoS-aware packet scheduling methods, which will otherwise try to compensate for inactive users with no data to transmit [17].

The Proportional Fair (PF) algorithm is presented in [18] and also analyzed in several other papers (see, for example, [19, 20]). The PF scheduler provides a trade-off between fairness and achievable cell throughput, thus providing significant coverage extension over the Max-C/I (maximum throughput) scheduler. It aims to serve users under very favorable instantaneous radio conditions relative to their average ones, thus taking advantage of the temporal variations of the fast fading channel. This relation is popularly interpreted as “scheduling users on top of their fades.”

6.4.1.3 Fast Fair Throughput Scheduling

Fast fair throughput scheduling is also known as Proportional Fair Throughput (PF-T) in [21]. Like the Proportional Fair method, this scheduling method is aimed at distributing the cell throughput fairly among the users while taking advantage of the short-term fading variations of the radio channel. In [22], a fast fair throughput algorithm that modifies the PF algorithm to

provide equalization of user throughput is proposed by Barriac as follows:

$$P_i = \alpha \frac{r_i(t)}{R_i} \; i = 1, \ldots, \; K$$

where α is the term that weights the PF algorithm to equalize the user throughputs and is given by

$$\alpha = \frac{\max j\{\overline{Rj}\}}{Rj}$$

where max $j\{\overline{Rj}\}$ is a constant that indicates the maximum average supportable data rate from all users, and $\overline{Rj}$ is the average supportable data rate of user i.

6.4.2 Slow Scheduling Methods

The slow scheduling methods are less aggressive than the aforementioned fast scheduling methods. They include scheduling algorithms that base their scheduling decisions on the average user's signal quality (e.g., the average C/I scheduler) or those that schedule without any consideration for user channel quality (e.g., "blind" Round Robin scheduler).

6.4.2.1 The Round Robin Scheduler

Round Robin scheduling is a "blind" scheduling technique that serves users in a cyclic manner without taking their radio channel conditions into consideration. Round Robin is also "fair time" scheduling; this means that the same power is allocated to all users such that a higher throughput is experienced by users with better channel conditions. This kind of scheduler is noted for its simplicity and assurance of a fair resource distribution among the users in the cell. Because Round Robin scheduling decision is oblivious to channel conditions, it typically suffers from low average throughput.

6.4.2.2 The Average C/I Scheduler

This scheduler selects, in every TTI, the user with the largest average C/I that has buffered data ready for transmission. An averaging window length of, for example, 20 to 100 ms is used such that only the user with maximum slow-averaged channel quality/throughput is served [13, 21].

6.4.2.3 The Fair Throughput Scheduler

The fair throughput scheduling approach aims at ensuring that all simultaneously queued users receive the same average throughput, which means

that users in bad channel conditions receive relatively more allocation of HS-DSCH resources. Thus, in every TTI, the user with the lowest average throughput is scheduled. This scheduler is considered a slow scheduling method because it does not employ instantaneous channel quality information.

6.4.3 Delay Differentiated Packet Scheduling Schemes

The previously discussed basic channel-aware and non-channel-aware scheduling algorithms do not explicitly take packet delays into account in their scheduling metrics. To provide QoS differentiation and to meet the delay requirements of real-time services, such as real-time streaming audio/video, conversational voice, and interactive video, researchers have proposed various scheduling algorithms that take packet delay into account for HSDPA MAC-hs scheduling. A few of these are discussed here and include Channel-Dependent Earliest Due Deadline (CD-EDD) [23], the Exponential Rule (ER) [22], and the Modified Largest Weighted Delay First (M-LWDF) [15]. These algorithms extend the opportunistic PF scheduling algorithm to include queuing delay metrics, thus allowing for delay differentiation between user packet flows.

6.4.3.1 Channel-Dependent Earliest Due Deadline

The CD-EDD algorithm is a combination of PF scheduling and an earliest deadline due (EDD) component resulting in the scheduling metric [23]:

$$P_i = w_i \frac{r_i(t)}{R_i} \frac{W_i(t)}{T_i - W_i(t)} \quad i = 1, \ldots, \; K$$

where w_i is a weighting factor for the user i, and $W_i\,(t)$ denotes the waiting time of the packet at the head of the queue (i.e., the HOL (head of line) packet). T_i is the maximum allowable delay for ith user, while $T_i - W_i(t)$ indicates the time until the deadline is reached. Thus, as time until deadline draws closer (i.e., when HOL packet delay approaches T_i), the EDD term dominates the scheduling metric, thereby giving the user a higher scheduling priority. On the other hand, when HOL packet delay is low, the user gets a low scheduling priority due to the EDD term.

6.4.3.2 Exponential Rule

The exponential rule (ER) algorithm strives to equalize the weighted delays of the queues of all flows if their differences happen to be large; but, for regular situations, the PF algorithm dominates. It has the following scheduling

metric [22]:

$$P_i = w_i \frac{r_i(t)}{R_i} \exp\left(\frac{w_i W_i(t) - \overline{wW(t)}}{1 + \sqrt{\overline{wW(t)}}}\right) \quad i = 1, \ldots, K$$

w_i is given by $w_i = -\log(\delta_i)/T_i$ with δ_i being the largest probability with which the scheduler may violate its deadline. $\overline{wW(t)}$ is defined as

$$\frac{1}{K} \sum\nolimits_k w_i W_i(t)$$

6.4.3.3 Modified Largest Weighted Delay First

The modified largest weighted delay first (M-LWDF) algorithm not only takes advantage of the multi-user diversity available in the shared channel available through the PF algorithm, but also increases the priority of flows with HOL packets close to their deadline violation. The scheduling metric is as follows. In each time slot, serve the queue j for which

$$w_i W_i(t)(r_i(t)/R_i)$$

is maximal, where $W_i(t)$ is the HOL packet delay for queue i, $r_i(t)/R_i$ is the PF term, and w_i is given by $w_i = -\log(\delta_i)/T_i$.

Thus, the greater the user i current packet delay, the channel quality relative to its average level, and the higher the QoS requirement (set through weights w_i), the greater the chance of that user being scheduled. One key feature of the M-LWDF algorithm is that a scheduling decision depends on both current channel conditions and the states of the queues. M-LWDF has proved to be throughput optimal in [24]; furthermore, it remains throughput optimal if for all or some users, the delay $W_i(t)$ is replaced by the queue length (amount of data) $Q_i(t)$ [15]. Because the M-LWDF algorithm only needs to time stamp the arriving data packets of all users, or keep track of the current queue length, it is quite easy to implement.

6.4.4 Scheduling with Inter-class Prioritization for End-User Multiplexed Diverse Flows

The above considered algorithms, while being able to provide delay differentiation, also possess metrics for *per-user* scheduling decisions only; the scheduling metrics do not include provisions for class differentiation between flows when an end user receives multiple flows with diverse QoS requirements in the same session. To address this problem, proposals based on modifications of the existing algorithms have been investigated.

For example, Golaup et al. [25] have modified the M-LWDF algorithm to allow for inter-class prioritization in addition to inter-user prioritization, thus supporting class differentiation between flows of the same user. The proposed algorithm is termed the Largest Average Weighted Delay First (L-AWDF) [26].

6.4.4.1 The Largest Average Weighted Delay First (L-AWDF)

The basic idea behind L-AWDF is that it maintains a PF factor for fairness with respect to channel conditions while giving transmission priority to those users whose scheduled packets are nearing the maximum allowable delays for their class through a relative delay factor; this is constructed in such a way as to allow traffic class performances to be influenced by different weighting factors. Note that in this algorithm, the priority of each class of traffic/flow is expressed through the specification of a *maximum allowable delay* and a selected *weighting factor*. The prioritization function for L-AWDF is defined as follows [25]:

$$P_i = \frac{R_i(t)}{\lambda_i(t)} \cdot \sum_{j=1}^{n} \left(\alpha j \cdot \frac{D_i,\, j(t)}{T_i,\, j} \right), \quad \sum_{j=1}^{n} \alpha_j = 1$$

where α_j is the weighting factor for traffic class j, and $D_{i,j}(t)$, $T_{i,j}$ are the observed head-of-line and maximum permissible delays, respectively, for traffic class j at user i. P_i denotes the user priority, $R_i(t)$ is the instantaneous data rate that can be supported by the user in the current TTI if served by the packet scheduler, and $\lambda_i(t)$ is the average throughput experienced by user i. Performance analyses of L-AWDF algorithm can be found in [25–27].

6.5 Buffer Management-Based Scheduling Approaches

The 3GPP HSDPA specifications do not define specific buffer management schemes for Node B operation, therefore allowing for network performance optimization through operator-specific implementation. However, employing buffer management schemes in the MAC-hs not only improves resource utilization at the air interface and efficiency at upper protocol layers, but also allows for customized QoS control, especially for multiple diverse flows transmitted to an end user in the same HSDPA session. Furthermore, with MAC-hs buffer management, end-to-end traffic performance gains can be achieved because of the queued flows at the air interface, which present a bandwidth bottleneck and unpredictable time-varying channel quality.

As mentioned previously, the packet scheduler's function is to schedule all UEs within the cell as well as to service priority queues, that is,

priority handling (see Figure 6.5). The scheduler schedules MAC-hs SDUs (generated from one or more queued MAC-d PDUs) based on information from upper layers. One UE may be associated with one or more MAC-d flows (priority queues). Each MAC-d flow contains HS-DSCH MAC-d PDUs for one or more priority queues. Hence, the buffer management algorithm (BMA) can be viewed as a subfunction of the MAC-hs scheduler that oversees the responsibility of handling priorities for multiple flows (priority queues) associated with one UE in the MAC-hs. With this approach, classical packet scheduling algorithms such as Round Robin, Max-C/I, PF, and M-LWDF can be applied to select the user eligible to receive transmission in the current TTI (inter-user transmission scheduling) while the BMA decides between the multiple flows/priority queues associated with the selected user (inter-class transmission prioritization) according to the BMA transmission prioritization policy. This approach works very well because a robust and efficient BMA can be designed to operate offline, while the packet scheduler performs the online task of scheduling the users in the cell.

The BMA performs buffer admission control (BAC) on arriving MAC-d PDUs according to the priority scheme's BAC mechanism. The BMA also determines the MAC-d flow (identifies the UE priority Queue ID), from which the MAC-d PDUs will be assembled into MAC-hs SDUs for transmission to the UE on a given HARQ entity. If priority switching is enabled [in the case of a dynamic buffer management scheme such as D-TSP (dynamic time-space priority) discussed later], the priority switching algorithm is consulted to identify the correct MAC-d flow (Queue ID) for next transmission. The scheduler then schedules the MAC-d flow on a selected HARQ process, if the scheduling algorithm has allocated the next transmission opportunity to the UE associated with the MAC-d flow. Thus, the BMA as a subentity of the MAC-hs scheduler, is executed on each UE's MAC-d flows, that is, the *priority handling* for UE's with multiple flows per session [e.g., multiplexed RT (real-time) and NRT (non-real-time) flows concurrent in the multimedia connection]. The entire scheduling process (see Figure 6.10) for multi-flow sessions therefore can be viewed as occurring in a logical hierarchy with the BMA performing buffer admission control and inter-class (inter-flow) scheduling, while MAC-hs scheduling performs the inter-UE scheduling according to any of the selected scheduling disciplines (e.g., Max-C/I, PF, or M-LWDF).

6.5.1 Integrated Scheduling and Buffer Management for HSDPA Multimedia Traffic

The queuing of packets in the Node B MAC-hs provides an opportunity to incorporate buffer management into the MAC-hs packet scheduling functionality in order to improve end-to-end traffic performance and resource

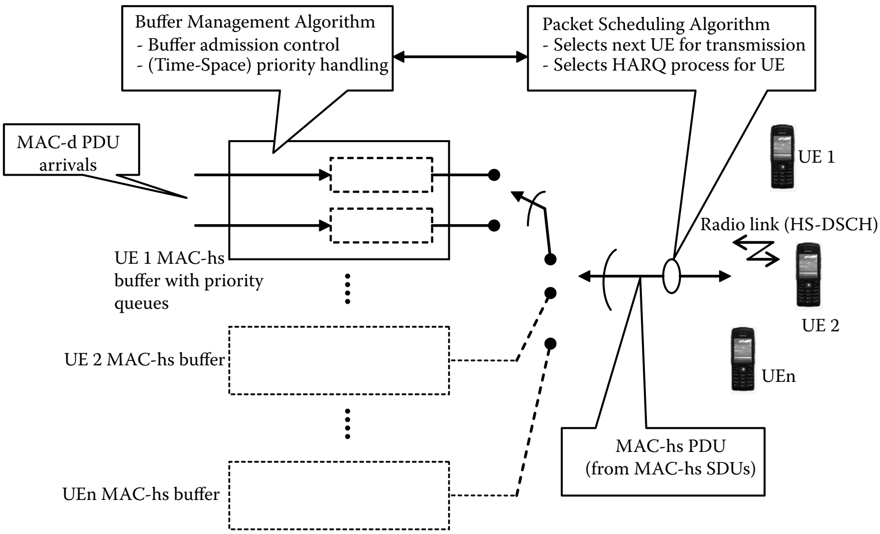

Figure 6.10 Integrated scheduling and buffer management in HSDPA. Logical hierarchy of UE_1 multiplexed RT and NRT MAC-d flows in Node B MAC-hs.

utilization. With higher data rates and improved downlink packet-switched services support enabled by HSDPA, multimedia applications, i.e., those with concurrent diverse "media" or data flows with different QoS requirements being multiplexed in the same end-user session, are a possible communication scenario for example, voice/video chatting with simultaneous data/file download. To schedule such end-user multimedia traffic in the MAC-hs, inter-user scheduling that also allows for inter-class transmission prioritization is necessary because the diverse media belong to different traffic classes that have different QoS requirements. The real-time voice/video components of the multimedia flow are partially loss tolerant while requiring low delay/jitter. On the other hand, the NRT components of the multimedia flow are delay tolerant but sensitive to packet loss. Thus, with the conflicting QoS requirements of the multiplexed components of the multimedia flow belonging to the same user, integrating a buffer management scheme with packet scheduling algorithms provides an effective mechanism for QoS control and performance optimization. The incorporated buffer management scheme must, of course, be able to prioritize the RT class for minimum queuing delay/jitter in the MAC-hs buffer, while at the same time minimizing packet loss for the NRT class. A buffer management scheme known as the time-space priority (TSP) has been proposed for inter-class priority handling and QoS control between multimedia session components (RT and NRT classes), in conjunction with inter-UE scheduling as shown in Figure 6.10. Thus, TSP (as an inter-class BMA) provides

per-session buffer management for the priority queues/flows associated with the multimedia user in the MAC-hs, by allowing space priority for the NRT flow while according time (transmission) priority to the RT flow.

6.5.1.1 Time-Space Priority Buffer Management

TSP [28] buffer management is a priority queuing scheme designed for joint QoS control of concurrent RT and NRT flows in an end-user multimedia session. Unlike most priority queuing that provides either delay or loss differentiation, TSP combines both loss and delay differentiation in a single (logical) queue, thus yielding transmission (time) priority for RT packets and space priority for NRT packets.

A threshold $\boldsymbol{R}$ is used to partition the queue, as shown in Figure 6.11, such that NRT packets are accorded a larger buffer space because of their loss sensitivity. $\boldsymbol{R}$ is a soft threshold that limits the maximum number of RT packets admitted into the queue at any given time to $\boldsymbol{R}$. Due to delay/jitter sensitivity of RT packets such as voice/video packets, they are queued on a first-come-first-served basis in front of the delay-tolerant NRT packets, which are also queued in a first-come-first-served manner behind the RT packets. However, the NRT packets enjoy unlimited access to the buffer such that the maximum number of admitted NRT packets at any given time can vary from $\boldsymbol{N} - \boldsymbol{R}$ to $\boldsymbol{N}$, where $\boldsymbol{N}$ is the total queue capacity. This allows for more efficient buffer space utilization and further minimization of NRT packet loss.

By virtue of TSP queuing, when the packet scheduler selects the UE with multimedia traffic for transmission in the current TTI, the transmission order of the multimedia flow components in TSP is RT packets first, and then NRT packets are allowed to be transmitted when there are no RT packets present in the queue/buffer. This implies static prioritization of real-time flow/class over the non-real-time flow/class. The restriction of admitted RT packets

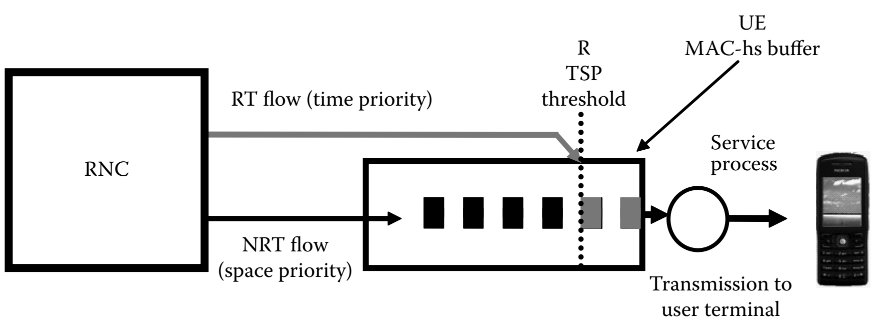

Figure 6.11 Time-Space Priority queuing for end-user multimedia traffic priority queuing and QoS control in the Node B MAC-hs packet scheduling entity.

into the queue also provides a mechanism to ensure that the non-time-flow does not get starved of transmission bandwidth especially at high real-time streams arrival intensity.

Due to loss sensitivity, the arrival of NRT packets at the RNC necessitates the use of RLC (Radio Link Control) Acknowledged Mode (AM) for onward transmission over the Iub interface to the Node B. RLC AM packets require feedback from a peer RLC entity in the UE, which is typically sent via a STATUS message. NRT packets that are lost or discarded from the Node B MAC-hs due to buffer overflow or excessive queuing delays can only be recovered with the RNC-based RLC (selective repeat) ARQ retransmissions. Because the ARQ mechanisms operate between the RNC and recipient UE, retransmissions increase the RLC round-trip-time, resulting in overall end-to-end delay of the NRT packets, which manifests in severe degradation of end-to-end-throughput for TCP-based NRT flow within a multimedia session. This is undesirable because the majority of NRT traffic utilizes TCP as the transport protocol. Furthermore, retransmissions lead to waste of Iub resources and Node B buffer space, as well as air interface bandwidth. The aforementioned motivates the utilization of Iub flow control to regulate MAC-hs queues of the multimedia sessions to allow efficient buffer and transmission resource utilization. An enhanced TSP buffer management scheme (E-TSP) proposed and evaluated in [11, 29] uses the Iub credit-based flow control mechanism discussed earlier to regulate the flow of the NRT component of the multimedia traffic in order to enhance end-to-end performance.

The E-TSP scheme is depicted in Figure 6.12. In addition to the TSP threshold R, it has additional thresholds ***L*** and ***H*** and a *credit allocation algorithm* designed to enable more efficient utilization of buffer space and air interface resources with minimal Iub signaling load. The allocation algorithm utilizes NBAP (Node B Application Part) signaling [6] as shown in the bottom part of Figure 6.12, to issue credits to the RNC that determine the number of arriving packet data units (PDUs) to the Node B for each flow in the user's session.

The number of credits per transmission interval is determined as follows [11]. Let the total credits per TTI for a particular multi-flow/multimedia user be given by

$$C_{\text{Total}} = C_{\text{NRT}} + C_{\text{RT}}$$

where

$$C_{\text{RT}} = (\lambda_{\text{RT}}/\text{PDU\_size}) \cdot \text{TTI}$$

C_{RT} is the number of credits per TTI for the RT flow in the multimedia session, while PDU_size denotes the size of the PDU in bits. λ_{RT} is the minimum guaranteed bit rate of the RT flow (in bps), a parameter that can

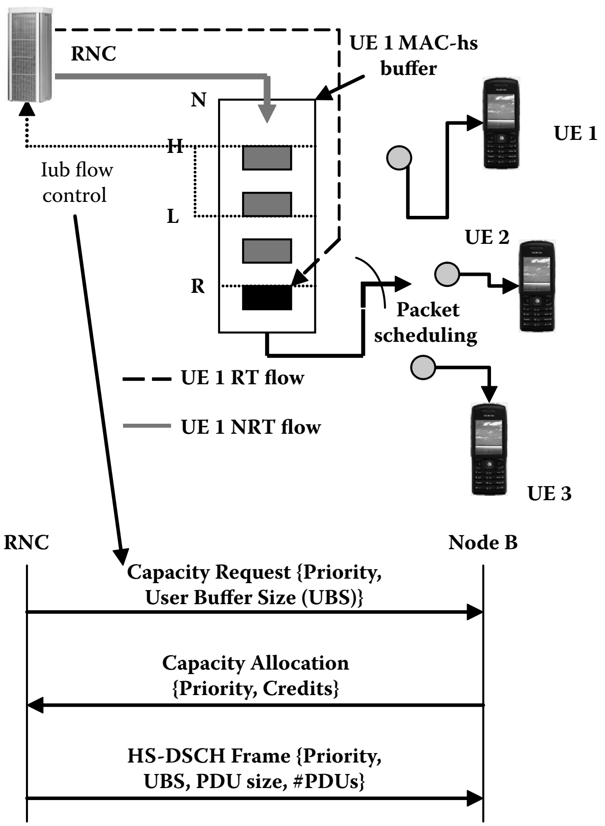

Figure 6.12 Enhanced Time-Space Priority (E-TSP) buffer management for per-UE multimedia traffic QoS control.

be obtained from the QoS attributes of the flow during bearer negotiation. For the NRT flow, the number of credits C_{NRT} is given by

$$C_{\mathrm{NRT}} = \min\{C_{\mathrm{NRTmax}}, \mathrm{UBS}_{\mathrm{NRT}}\}$$

where $\mathrm{UBS}_{\mathrm{NRT}}$ is the user's NRT buffer occupancy in the RNC and C_{NRTmax} is the maximum NRT grants per TTI, which depends on the HSDPA channel load, scheduling policy, and the recipient UE radio conditions. C_{NRTmax} is calculated from

$$C_{\mathrm{NRTmax}} = (\lambda'_{\mathrm{NRT}}/\mathrm{PDU\_size}) \cdot \mathrm{TTI}, \ \boldsymbol{N_T} < \boldsymbol{L}$$

$$k \cdot (\lambda'_{\mathrm{NRT}}/\mathrm{PDU\_size}) \cdot \mathrm{TTI}, \ \boldsymbol{L} \leq \boldsymbol{N_T} \leq \boldsymbol{H}$$

$$0, \ \boldsymbol{N_T} > \boldsymbol{H}$$

where $\boldsymbol{L}$ and $\boldsymbol{H}$ are the flow control thresholds in Figure 6.12, and $\boldsymbol{N}_T$ is the total number of RT and NRT PDUs in the queue. $0 < k < 1$ is a factor for overflow control and λ'_{NRT} is an estimate of the user's NRT data rate allocated by the packet scheduler in the MAC-hs. The estimate is obtained using an exponentially weighted moving average filter according to

$$\lambda'_{\mathrm{NRT}} = \alpha \cdot \lambda'_{\mathrm{NRT}-1} + (1 - \alpha) \cdot \lambda_{\mathrm{NRT}}$$

where λ_{NRT} is the instantaneous NRT bit rate. With the expression for C_{NRTmax}, NRT grant allocation is made dependent of load and user channel quality, which is appropriate because of the elastic nature of the NRT flow. Because averages are used in the grant calculation, the space between $\boldsymbol{H}$ and $\boldsymbol{N}$ absorbs instantaneous burst arrivals.

6.5.1.2 Buffer Management with Dynamic Inter-Class Prioritization

Although the TSP and E-TSP schemes can be utilized as the BMA to allow inter-class QoS control for integrated scheduling and buffer management solution in the MAC-hs, the static transmission prioritization of real-time class in multimedia flow may not always yield the optimum QoS. Within the BMA, *dynamic* transmission prioritization will allow the delay flexibility and partial loss tolerance of the RT streams to be exploited to optimize bandwidth allocation to the NRT streams. This will improve end-to-end performance as well as further mitigate potential NRT streams starvation in the multimedia session. In [30], dynamic time-space priority (D-TSP) is proposed as an enhancement to E-TSP to enable optimized QoS control of the RT and NRT streams of the multimedia session. The end-to-end performance evaluation of D-TSP using extensive system-level HSDPA simulations can also be found in [30].

D-TSP as shown in Figure 6.13 incorporates dynamic switching of time (transmission) priority between the RT and NRT streams present in the multimedia user's MAC-hs buffer.

For a given transmission opportunity assigned to the UE with multimedia traffic by the (inter-UE) packet scheduling algorithm, when there is no danger of HOL queuing delay of multimedia real-time packets exceeding a given delay budget, the transmission priority is switched to the NRT flow. If real-time HOL delay is greater than or equal to the delay budget or no non-real-time packets are present in the D-TSP queue, then transmission priority remains with the real-time flow. The delay budget can be expressed in terms of the number of queued real-time packets via a parameter $\boldsymbol{k}$, where [30]

$$\text{Delay budget} = \boldsymbol{k} \times (\text{RT packet inter-arrival time})$$

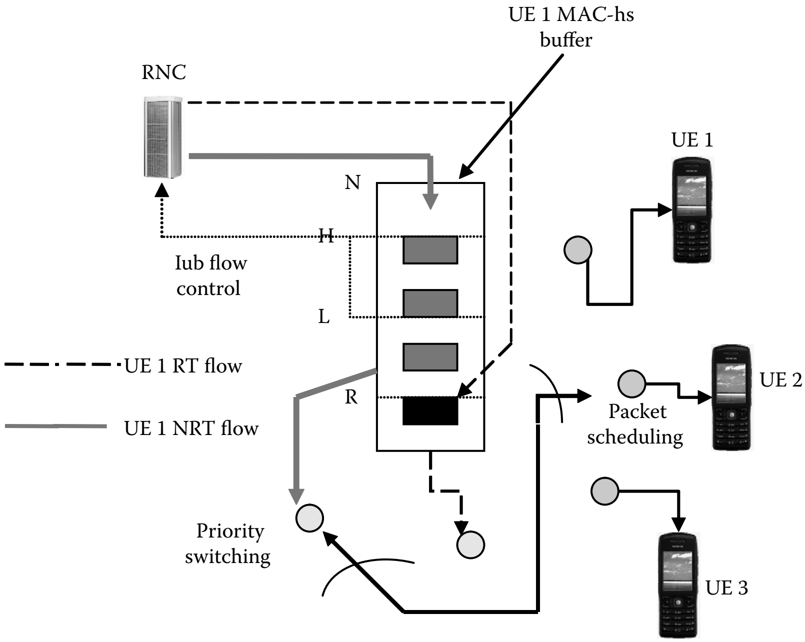

Figure 6.13 Enhanced TSP with dynamic priority switching (D-TSP).

Thus, $\boldsymbol{k} = 2$ with a real-time packet inter-arrival time of 20 ms is equivalent to a delay budget of 40 ms.

Let *MAX_delay* represent the maximum allowable queuing delay to enable end-to-end QoS delay guarantee for the RT streams in the multimedia flow. A discard timer (DT) is set on arrival of real-time packets (PDUs) to the MAC-hs buffer. DT is configured to time-out after a period of *MAX_delay,* triggering the dropping of HOL RT packet(s) queued for up to *MAX_delay* seconds. DT is cancelled on transmission of RT packet(s). We can therefore express the time priority switching strategy as follows:

IF RT packets < **k** AND RT HOL delay < *MAX_delay* AND
NRT packets > 0
Time Priority = NRT flow
Generate Transport Block from NRT PDUs
ELSE
Time Priority = RT flow
Generate Transport Block from RT PDUs

D-TSP provides an effective buffer management algorithm for an integrated scheduling and buffer management solution in HSDPA because it ensures

that the NRT stream is allocated the optimal bandwidth at the air interface within the allowable delay tolerance of the companion RT stream in the multimedia session of the user, while the allocation of transmission slots between the multimedia user and other users in the cell is done via an implemented inter-user packet scheduling algorithm such as PF, M-LWDF, or Max-C/I, depending on operator requirements.

6.6 Summary

This chapter provided detailed coverage of physical and MAC layer mechanisms in the 3GPP standards that enable High-Speed Downlink Packet Access (HSDPA) in WCDMA UMTS systems. This provided a basis for our discussion of packet scheduling and buffer management solutions in HSDPA. Packet scheduling is embedded in a new MAC-hs entity at Node B close to the air interface. The MAC-hs also incorporates credit-based flow control algorithms to facilitate HS-DSCH congestion control. The MAC-hs entity in an HSDPA Node B is also equipped with priority handling functionality to manage resources between HARQ entities, which could consist of multiple queues for flows associated with a single user. Thus, the HSDPA standards have existing mechanisms to facilitate buffer management for prioritizing multiple flows with different traffic class requirements in an ongoing, end-user multimedia session. This allows integrated scheduling and buffer management solutions to be employed for enhanced end-to-end QoS performance of real-time and non-real-time classes of flows in the same end-user multimedia sessions.

In HSDPA, opportunistic or channel-aware packet schedulers exploit multi-user diversity to increase the gain in total cell throughput usually at the expense of fairness. Nevertheless, the conflicting goals of maximizing throughput while ensuring some degree of fairness of users requesting service is a major design consideration of HSDPA scheduling algorithms. While basic schedulers such as the Maximum C/I, Round Robin, and Proportional Fair do not explicitly account for packet delays in their scheduling metrics, some proposed algorithms such as M-LWDF and CD-EDD provide delay differentiation to allow the delay requirements of real-time services to be met.

To allow packet scheduling with inter-class prioritization between classes or media in a multimedia session of the same user, algorithms such as the L-AWDF have been proposed. L-AWDF allows service differentiation through an expressed maximum allowable delay and a weighting factor for each class of the user's multimedia traffic. An alternative approach to scheduling with inter-class prioritization, which was also discussed, is the integrated scheduling and buffer management approach. This uses a buffer management scheme for inter-class prioritization within the same

user multimedia flows while the packet scheduling discipline selects the eligible user for transmission. We presented a TSP buffer management scheme as an effective per-user algorithm that can be used jointly with existing inter-user packet scheduling disciplines for multimedia sessions with real-time and non-real-time classes of flows. An enhanced version (E-TSP) leverages the HSDPA flow control to enhance end-to-end performance. A dynamic version (D-TSP) that allows adaptive inter-class priority switching between the real-time and non-real-time flows, thereby optimizing the QoS control between the end-user multimedia session flows, was also described.

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